CALL ROUTING USING INFORMATION IN SESSION INITIATION PROTOCOL MESSAGES

CROSS-REFERENCE TO RELATED APPLICATION(S)

This application claims the benefit of U.S. provisional application number 60/281,143 filed on April 3, 2001, the content of which is incorporated herein by reference. This application also contains subject matter which is related to the subject matter disclosed in U.S. application entitled "Session Initiation Protocol Routing Using Voice Cookies" (attorney docket (47339/JEC/X2), filed on December 5, 2001, the content of which is also incorporated herein by reference.

FIELD OF THE INVENTION

This invention relates generally to Internet telephony systems, and more particularly, to intelligent call routing in Internet telephony systems that adhere to session initiation protocol.

BACKGROUND OF THE INVENTION

Session Initiation Protocol (SIP) is a signaling protocol for creating, modifying, and terminating multimedia sessions, including Internet telephone calls, with one or more SIP-endpoints. Details about the SIP signaling protocol is set forth in Internet Engineering Task Force Request for Comment 2543 entitled "SIP: Session Initiation Protocol," March 1999 (hereinafter referred to as RFC 2543), which is incorporated herein by reference. SIP provides an alternative to PBX- or H.323-signaled telephony.

Although SIP end-points can directly place calls to one another, SIP servers, including proxy and redirect servers, are typically engaged during the call set-up process to route calls. Such call routing includes ascertaining called end-points in response to call establishment messages, referred to as INVITE messages, originated by calling end-points. The INVITE messages are either proxied to ascertained called end-points or the addresses of ascertained called end-points are returned to the calling end-points for allowing the calling end-points to directly place calls to the called end-points.

FIG. 1A is a functional block diagram for establishing a SIP call via a typical proxy SIP server 10. In step 30, the proxy server 10 receives an invitation from a calling end-point 15 in the form of an INVITE request. The INVITE request includes routing information in the "From:", "To:", "Contact:" and other standardized fields within the INVITE message header. The "To:" field of the message header includes a generic SIP URL associated with a called end-point 20.

The proxy server 10 accepts the INVITE request and in step 32, preferably engages a location server 25 for routing the call based on the routing information in the SIP message header. In this regard, the location server 25 retrieves the SIP URL associated with the called end-point to resolve the URL to a more precise address. For example, a call directed to a generic SIP URL such as, for example, "sales@acme.com" may be resolved to a particular person, such as, for example, "bob@ny.acme.com." The retrieved address information is transmitted to the proxy server 10 in step 34.

In step 36, the proxy server 10 issues a second INVITE request to the more precise address. The called end-point 20

receives the second INVITE request and alerts the user of the request by, for example, causing the user's telephone to ring. If the call is answered, the called end-point 20, in step 38, returns a success indication to the proxy server 10 via an OK response. The proxy server 10 forwards the OK response in step 40 to the calling end-point 15. The receipt of the success result is confirmed by the calling end-point 15 by transmitting an ACK request to the proxy server 10 in step 42, which then forwards it to the called end-point 20 in step 44.

FIG. 1B is a functional block diagram of an alternative method for establishing a SIP call using a typical redirect SIP server 47. In step 31, the redirect server 47 accepts the INVITE request and, as the proxy server 10 of FIG. 1A, contacts the location server 25 in step 33 for routing the call based on the routing information in the INVITE message header. The redirect server 47, instead of directly contacting the newly found address received in step 35, returns the address to the calling end-point 15 in step 37. The calling end-point 15 confirms receipt of the address via an ACK request in step 39.

The calling end-point 15 issues a new INVITE request to the address returned by the redirect server 30 in step 41. If the call succeeds, the called end-point 20 transmits an OK response in step 43, and the calling end-point 15, in step 45, completes the handshake with an ACK request.

One limitation in current SIP call routing is the limited information on the caller's intent that may be deduced at the time the call is initiated from standard routing fields within the INVITE message headers. In order to gather additional call intent information for routing a call, conventional approaches often make use of interactive voice response (IVR) systems,

whereby the caller is prompted for and provides additional information on the caller's intent through selection of dual tone multi-frequency (DTMF) digits on the telephone keypad. For example, a person making a call to a general address may be asked to enter account information through use of DTMF telephone keypad digit entry, and select a particular department, such as customer service, sales, or marketing department. Only after this information is entered can the call be appropriately routed to a suitable call center operator. Use of IVR systems to ascertain additional caller intent information is very cumbersome and inconvenient for the caller, and requires additional message exchanges and database lookups, which translate into slow call setup times and frustration for the caller.

Conventional telephony systems may employ caller ID data that is automatically transmitted with an incoming call in lieu of DTMF key entries to route the call. However the data transmitted is limited to caller ID data, and does not include additional caller intent information that may be desirable to more intelligently route a call.

Newer systems may employ voice recognition techniques in response to IVR prompts to deduce the caller's intent. However, such voice recognition systems are also cumbersome and inconvenient for the caller, subject to error, and also yield slow call setup times.

Accordingly, there is a need for a more efficient system and method for ascertaining caller intent information for intelligent and efficient routing of incoming calls.

SUMMARY OF THE INVENTION

The current invention is directed to a system and method for intelligently routing IP telephone sessions using information transmitted by the caller that extends beyond simple caller ID data. In one embodiment, the invention is directed to a method for establishing an IP telephony session between a first device and a second device where the method includes receiving a call establishment message from the first device, the call establishment message including a first routing information in a header portion of the message, and a second routing information in a body portion of the message. The method further includes determining an address of the second device based on the first and second routing information and using the address for routing the session to the second device.

In another embodiment, the invention is directed to a method for establishing an IP telephony session between a first device and a second device where the method includes retrieving caller intent information from a data store on the first device, creating a call establishment message including a header and a body, the body including the caller intent information, and transmitting the caller initiation request to a server for routing the session to the second device based on the caller intent information.

In a further embodiment, the invention is directed to an IP telephony system including a first device, a second device, and a server operative between the first device and the second device. The first device creates a call establishment message including a header and a body, the body including caller intent information. The server uses the caller intent information for

determining an address of the second device for routing the session to the second device.

In another embodiment, the invention is directed to a server in an IP telephony system operative between a first device and a second device. The server is configured for receiving a call establishment message from the first device, the call establishment message including a first routing information in a header portion of the message and a second routing information in a body portion of the message, determining an address of the second device based on the first and second routing information, and using the address for routing the session to the second device.

It should be appreciated, therefore, that the present invention allows a more intelligent determination of a callee address than when only using standard routing information typically included in the header portion of a call establishment message or traditional caller ID information. Additional information about the caller included in the body portion of the call establishment message allows an efficient determination of the caller's intent for routing the call to a most appropriate callee. Because the caller intent information is already made available at the placement of the call, back and forth interactions with the caller for obtaining such information is no longer required.

BRIEF DESCRIPTION OF THE DRAWINGS

These and other features, aspects and advantages of the present invention will be more fully understood when considered with respect to the following detailed description, appended claims, and accompanying drawings where:

FIG. 1A is a functional block diagram of establishing a SIP call using a typical proxy SIP server;

FIG. 1B is a functional block diagram of an alternative method for establishing a SIP call using a typical redirect SIP server;

FIG. 2 is a schematic block diagram of a communication system adhering to an IP telephony protocol according to one embodiment of the invention;

FIG. 3 is a functional block diagram for establishing a SIP call based on caller information according to one embodiment of the invention where a proxy SIP server is used;

FIG. 4 is a functional block diagram for establishing a SIP call based on caller information according to one embodiment of the invention where a redirect SIP server is used;

FIG. 5 is a functional block diagram for determining an address of a called end-point using caller information according to one embodiment of the invention; and

FIG. 6 is a flow diagram of a process for an IP telephony call setup according to one embodiment of the invention.

DETAILED DESCRIPTION

FIG. 2 is a schematic block diagram of a communication system adhering to an IP telephony protocol such as, for example, a session initiation protocol (SIP), according to one embodiment of the invention. The system includes a calling end-point 50 initiating a call that is directed to a called end-point 52 over a wide area network, such as, for example, a public internet 54. The calling end-point 50, called end-point 52, and internet 54 preferably adhere to the SIP signaling protocol set forth in RFC 2543. A person skilled in the art should recognize, however,

that the any other IP signaling protocol conventional in the art may be used instead of SIP.

The calling and called end-points 50, 52 are preferably SIP-enabled telephones, hand phones, personal computers, switches, routers, and/or the like. Preferably, each calling and called end-point is associated with an input device 59, 65 receiving input data, such as, for example, an input cable, a keyboard, a keypad, or the like. Each calling and called end-point is further associated with an output device 61, 63 presenting output data, such as, for example, an output cable, a display, or the like. The calling and called end-points 50, 52 are preferably also associated with handsets 60, 62 for allowing a voice conversation between a caller and callee.

The communication system of FIG. 2 further includes a server 56 preferably routing calls between the calling and called end-points 50, 52. The server 56 is preferably a SIP proxy or redirect server similar to the proxy and redirect servers 10, 47 of FIGS. 1A-1B. The server 56, however, is enhanced with the capability of using information transmitted by the calling end-point along with standard routing information, for more intelligently routing the call.

The server 56 is preferably coupled to a database 57 and location server 58. The location server contains location information used for routing calls between the calling end-point and the called end-point. The location server may be similar to the location server 25 of FIGS. 1A-1B. The database 57 is a data store, such as a hard disk drive or drive array, storing caller information and associated routing information for use in addition to the location information to route the calls.

According to one embodiment of the invention, the server 56, database 57, and/or location server 58 reside in a single machine. In another embodiment, the servers reside in two separate machines coupled to each other over a local area network, private wide area network, or the public internet 54.

In general terms, the calling end-point 50 initiates a call by transmitting a call initiation message to the server 56. The call initiation message may be, for example, a SIP INVITE message. The call initiation message includes caller information in addition to standard routing information that may be used by the server 56 to route the call. Such caller information preferably relates to the caller's intent, and may include, by way of example, a name, an address, a phone number, an email address, an account number, a transaction history, a billing history, a department name or selection, an agent preference or selection, a language preference or selection, a product preference or selection, or any other information that may be used to route a call, such as, for example, information typically gathered via an IVR system. The caller information may also include user profile information, user habit information, or information contained in traditional browser cookies associated with a web site domain.

The caller information is preferably generated and stored in the calling end-point 50 according to various mechanisms. For instance, the caller information data may be generated by completing an electronic form that may have a click-to-call button for initiating a call. The caller information may also be written by the server 56 or calling end-point 50 during or after a call. The caller information may, for instance, update

transaction history information, call history information, called agent information, and/or the like.

The server 56 receives the call initiation message from the calling end-point 50 and determines an address to which to route the call using information retrieved from the location server 58 and database 57. The call is then routed to the called end-point 52 based on the address, or the address returned to the calling end-point 50 for allowing it to establish a direct communication with the called end-point.

FIG. 3 is a functional block diagram for establishing a SIP call based on caller information according to one embodiment of the invention where the server 56 is a proxy server 56a. In step 70, the proxy server 56a receives an INVITE message from the calling end-point 50. The INVITE request preferably includes standard routing information in the "From:", "To:", "Contact:" and other fields within the INVITE message header, referred to as the header routing information (HRI). The INVITE request further includes caller information in the body of the INVITE message, referred to as the body routing information (BRI).

The proxy server 56a ascertains an address of the called end-point 52 based on the header and body routing information. For example, the header routing information may contain an address of a general appliance company to which the call is directed. The body routing information may contain keywords associated with the caller, such as, for example, "washing machine," "service," and "model ABC." Based on this information, the proxy server 56a may deduce the caller intent and route the call to an agent skilled in answering questions about the servicing of washing machine model ABC using information retrieved from the database 57 and location server 58.

In step 80, the proxy server 56a issues a second INVITE message to the ascertained address. All or portions of the received body routing information may also be provided to the called end-point in the body of the second INVITE message. This may be desirable, for example, to facilitate discussions between a customer representative at the called end-point 52 and a caller at the calling end-point 50.

The called end-point 52 receives the second INVITE message and alerts the user of the request by, for example, causing the user's telephone to ring. If the call is answered, the called end-point 52, in step 82, returns a success indication to the proxy server 56a via an OK response. The proxy server 56a in turn forwards the OK response in step 84 to the calling end-point 50. The receipt of the success result is confirmed by the calling end-point 50 by transmitting an ACK request to the proxy server 56a in step 86, which then forwards it to the called end-point 52 in step 88. Voice conversation between users of the calling and called end-points 50, 52 ensues in step 90 via their respective handsets 60, 62.

FIG. 4 is a functional block diagram for establishing a SIP call based on caller information according to another embodiment of the invention where the SIP server 56 is a redirect server 56b. In step 100, the redirect server 56b accepts an INVITE message including header and body routing information, and, as the proxy server 56a for FIG. 3, examines the header and body information for ascertaining an address to the called end-point using information retrieved from the database 57 and location server 58.

In step 112, the redirect server 56b returns the ascertained address to the calling end-point 50 which confirms receipt of the

address via an ACK request. In step 114, the calling end-point 50 issues a new INVITE request to the address returned by the redirect server 56b. If the call succeeds, the called end-point 52 transmits an OK response, and the calling end-point 50 completes the handshake with an ACK request. Voice conversation between the users of the calling and called end-points 50, 52 ensues in step 116 via their respective handsets 60, 62. All or portions of the received body routing information may also be provided to the called end-point in the body of the second INVITE message. This may be desirable, for example, to facilitate discussions between a customer representative at the called end-point 52 and a caller at the calling end-point 50.

FIG. 5 is a functional block diagram for determining an address of a called end-point according to one embodiment of the invention. A location service 100 receives an INVITE message 102 from the calling end-point 50. The message preferably contains header routing information 102a preferably including "From:," "To:," "Contact:," and other standardized routing fields set forth in RFC 2543. In addition to the header routing information, the message preferably also contains body routing information 102b. The body routing information 102b includes information associated with the caller that may be used to deduce the caller's intent.

The location server 100 produces an address 106 as a function of the header routing information and body routing information. In this regard, the location server 100 may deduce the address using information in the location server 58 and database 57 and/or based on rules and processing logic for ascertaining a most appropriate address. The address may be, for instance, a specific IP address.

The location service is preferably a software module residing in the server 56. Alternatively, the location service may be a software module residing in the location server 58 and/or database 57. A person skilled in the art should recognize, however, that the location service may be implemented in firmware, hardware, or in any combination of software, firmware, and/or hardware.

FIG. 6 is a flow diagram of a process for an IP telephony call setup according to one embodiment of the invention. The process starts, and in step 210, the calling end-point 50 determines if a request was received to initiate a call. A caller may request that a call be initiated by, for example, selecting a click-to-call button from a web page provided by a browser resident in the calling end-point. If a request was received, the calling end-point searches a data store, in step 212, for the caller information. The data store may be a RAM, disk drive, or any other storage device conventional in the art. The calling end-point searches for a particular file-name, file extension, or the like, where caller information is known to be stored. Preferably, the calling end-point retrieves caller information that is relevant to the destination of the call and the associated server 56.

According to one embodiment of the invention, the retrieved caller information is used to fill entries in the web page for user verification. For example, the web page may provide entries for a last called department, a last called agent, a product name, an account number, and the like, which the calling end-point automatically fills out based on the retrieved caller information. The user may modify the automatically filled information after review.

In step 214, the calling end-point creates a call initiation request message, such as for example, a SIP INVITE message, with standard header routing information for transmitting to the server 56. In addition, the calling end-point, in step 216, includes the retrieved caller information in the body of the call initiation request message as body routing information.

In step 218, the calling end-point transmits the call initiation request message including the header and body routing information to the server 56. The server 56 receives the call initiation request message and retrieves from it the header routing information and the body routing information. In step 220, the server 56 determines an address of the called end-point as a function of the header routing information and the body routing information. For example, while the header routing information may indicate a general customer service number in the "To:" field, the body routing information may indicate a particular department or agent to which to route the call. In step 222, the SIP server routes the call to an end-point based on the determined address.

In step 224, a determination is made as to whether new caller information is to be written to the calling end-point 50. Such information may be provided by the server 56, called end-point 52, or the calling end-point itself. If the answer is YES, the calling end-point stores the information in an appropriate storage location. The information is then used for routing future body routing information.

Although this invention has been described in certain specific embodiments, those skilled in the art will have no difficulty devising variations which in no way depart from the scope and spirit of the present invention. It is therefore to

be understood that this invention may be practiced otherwise than is specifically described. Thus, the present embodiments of the invention should be considered in all respects as illustrative and not restrictive, the scope of the invention to be indicated by the appended claims and their equivalents rather than the foregoing description.